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**Information technology —  
Telecommunications and information  
exchange between systems — Corporate  
Telecommunication Networks —  
Signalling Interworking between QSIG  
and SIP — Call Diversion**

*Technologies de l'information — Télécommunications et échange  
d'information entre systèmes — Réseaux de télécommunications  
d'entreprise — Interaction de signalisation entre QSIG et SIP —  
Déviation d'appel*

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## Foreword

ISO (the International Organization for Standardization) and IEC (the International Electrotechnical Commission) form the specialized system for worldwide standardization. National bodies that are members of ISO or IEC participate in the development of International Standards through technical committees established by the respective organization to deal with particular fields of technical activity. ISO and IEC technical committees collaborate in fields of mutual interest. Other international organizations, governmental and non-governmental, in liaison with ISO and IEC, also take part in the work. In the field of information technology, ISO and IEC have established a joint technical committee, ISO/IEC JTC 1.

International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 2.

The main task of the joint technical committee is to prepare International Standards. Draft International Standards adopted by the joint technical committee are circulated to national bodies for voting. Publication as an International Standard requires approval by at least 75 % of the national bodies casting a vote.

Attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights. ISO and IEC shall not be held responsible for identifying any or all such patent rights.

ISO/IEC 23915 was prepared by Ecma International (as ECMA-360) and was adopted, under a special "fast-track procedure", by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, in parallel with its approval by national bodies of ISO and IEC.

## Introduction

This International Standard is one of a series of Standards defining the interworking of services and signalling protocols deployed in corporate telecommunication networks (CNs) (also known as enterprise networks). The series uses telecommunication concepts as developed by ITU-T and conforms to the framework of International Standards on Open Systems Interconnection as defined by ISO/IEC.

This International Standard specifies interworking between the Session Initiation Protocol (SIP) and QSIG within corporate telecommunication networks (also known as enterprise networks) for calls that undergo diversion. SIP is an Internet application-layer control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include, in particular, telephone calls. QSIG is a signalling protocol for creating, modifying and terminating circuit-switched calls, in particular telephone calls, within Private Integrated Services Networks (PISNs). QSIG is specified in a number of Standards and published also as ISO/IEC International Standards.

This International Standard is based upon the practical experience of member companies and the results of their active and continuous participation in the work of ISO/IEC JTC1, ITU-T, IETF, ETSI and other international and national standardization bodies. It represents a pragmatic and widely based consensus.

# Information technology — Telecommunications and information exchange between systems — Corporate Telecommunication Networks — Signalling Interworking between QSIG and SIP — Call Diversion

## 1 Scope

This document specifies signalling interworking between "QSIG" and the Session Initiation Protocol (SIP) in support of call diversion within corporate telecommunication networks (CN), also known as enterprise networks.

"QSIG" is a signalling protocol that operates between Private Integrated services Network eXchanges (PINX) within a Private Integrated Services Network (PISN). A PISN provides circuit-switched basic services and supplementary services to its users. QSIG is specified in Standards, in particular [1] (call control in support of basic services), [2] (generic functional protocol for the support of supplementary services) and a number of Standards specifying individual supplementary services. Diversion services are specified in [4] and the QSIG signalling protocol in support of these services is specified in [5]. In particular, this signalling protocol signals information about call diversion to the users involved.

SIP is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over IP [8], [10]. Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in [11]. An extension to SIP provides history information [14] that can be used to signal information about the retargeting of a request, in particular a call establishment request, as it is routed through a network.

This document specifies signalling interworking for call diversion during the establishment of calls between a PISN employing QSIG and a corporate IP network employing SIP. It covers both the impact on SIP of call diversion in the QSIG network and the impact on QSIG of request retargeting in the SIP network. Signalling interworking for call diversion operates on top of signalling interworking for basic calls, which is specified in [6].

Call diversion interworking between a PISN employing QSIG and a public IP network employing SIP is outside the scope of this specification. However, the functionality specified in this specification is in principle applicable to such a scenario when deployed in conjunction with other relevant functionality (e.g., number translation, security functions, etc.).

This specification is applicable to any interworking unit that can act as a gateway between a PISN employing QSIG and a corporate IP network employing SIP.

## 2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

[1] International Standard ISO/IEC 11572 "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Circuit mode bearer services — Inter-exchange signalling procedures and protocol" (also published by Ecma as Standard ECMA-143).

[2] International Standard ISO/IEC 11582 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Generic functional protocol for the support of supplementary services -- Inter-exchange signalling procedures and protocol" (also published by Ecma as Standard ECMA-165).

[3] International Standard ISO/IEC 13868 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Name identification supplementary services" (also published by Ecma as Standard ECMA-164).

[4] International Standard ISO/IEC 13872 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Specification, functional model and information flows -- Call Diversion supplementary services" (also published by Ecma as Standard ECMA-173).

[5] International Standard ISO/IEC 13873 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Call Diversion supplementary services" (also published by Ecma as Standard ECMA-174).

[6] International Standard ISO/IEC 17343 "Information technology -- Telecommunications and information exchange between systems -- Corporate telecommunication networks -- Signalling interworking between QSIG and SIP -- Basic services" (also published by Ecma as Standard ECMA-339).

[7] Ecma Technical Report TR/86, "Corporate Telecommunication Networks – User Identification in a SIP/QSIG Environment".

[8] J. Postel, "Internet Protocol", RFC 791.

[9] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119.

[10] S. Deering, R. Hinden, "Internet Protocol, Version 6 (IPv6)", RFC 2460.

[11] J. Rosenberg, H. Schulzrinne, et al., "SIP: Session initiation protocol", RFC 3261.

[12] J. Peterson, "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323.

[13] H. Schulzrinne, D. Oran, G. Camarillo, "The Reason Header field for the Session Initiation Protocol (SIP)", RFC 3326.

[14] M. Barnes "An Extension to the Session Initiation Protocol for Request History Information", draft-ietf-sipping-history-info-03 (work in progress).